providing a second communication device coupled to the Internet;

initiating a call using the first communication device to the second communication device using by establishing an initial Q.931 protocol;

D1 Cont.

establishing far end control of the second communication device by the first communication device in accordance with an H.245 protocol;

performing gatekeeper signaling in the first communication device accordance with an RAS protocol; and

packetizing audio information of the telephonic communication for transfer over the Internet using a standard real-time transfer protocol (RTP).

16. *(New)* An arrangement for providing telephonic communication that may be selectively transmitted via the Internet using standard Internet protocols, comprising:

a telephone; and

interface means coupled to the telephone and configured and arranged to receive audio information of the telephonic communication, the interface means comprising:

first output means configured to be coupled to a standard switched telephone communications network,

D2

second output means configured to be coupled to an Internet communications network, and

processing means configured and arranged to determine if the audio information received from the telephone is to be coupled to the first output means to establish a standard telephonic communication using the standard switched telephone communications network, or if the audio information is to be processed in accordance with the standard Internet transfer protocols and coupled to the second output means to establish an Internet communication using the Internet communications network to communicate the processed audio information in accordance with the standard Internet transfer protocols.

17. *(New)* A method for providing telephonic communication that may be selectively transmitted via the Internet using standard Internet protocols, the method comprising:

providing an interface unit having a memory and adapted to receive telephonic communication in response to user intervention and to communicate the telephonic communication via at least one of: a first output coupled to a standard switched telephone network and a second output coupled to an Internet communications network;

providing a telephone device communicatively coupled to the interface unit;

generating audio information at the telephone and sending the information to the interface unit;

automatically determining, at the interface unit, if the audio information received from the telephone is to be coupled to the first or second output; and

responsive to the determination, coupling the telephone via the interface unit to at least one of the standard switched telephone network and the Internet communications network.

18. *(New)* The method of claim 17, wherein automatically determining if the audio information is to be coupled to the first or second output is responsive to comparing a DTMF code received as part of the audio information to a variable stored in memory at the interface and is without further user intervention.

19. *(New)* The method of claim 17, wherein automatically determining if the audio information is to be coupled to the first or second output is responsive to detecting a DTMF code received as part of the audio information that represents the number for a local Internet access provider and is without further user intervention.

20. *(New)* The method of claim 17, wherein automatically determining if the audio information is to be coupled to the first or second output is responsive to comparing a DTMF code received as part of the audio information to a telephone number stored in memory at the interface and is without further user intervention.

21. *(New)* The arrangement of claim 1, wherein the interface unit further comprises a memory, and wherein the processing unit is adapted to automatically determine if the audio information is to be coupled to the first or second output by comparing a DTMF code received as part of the audio information to a variable stored in memory at the interface, without further audio information.

22. *(New)* The arrangement of claim 1, wherein the processing unit is adapted to automatically determine if the audio information is to be coupled to the first or second output by detecting if a DTMF code received as part of the audio information represents the number for a local Internet access provider, without further audio information.

23. *(New)* The arrangement of claim 1, wherein the interface unit further comprises a memory, and wherein the processing unit is adapted to automatically determine if the audio information is to be coupled to the first or second output by comparing a DTMF code received as part of the audio information to a telephone number stored in memory at the interface, without further audio information.

Remarks

Favorable reconsideration of this application is requested in view of the following remarks. For the reasons set forth below, Applicant respectfully submits that the claimed invention is allowable over the cited references.

The Office Action dated February 28, 2000, indicated that the information disclosure statement filed on June 25, 1998 fails to comply with 37 C.F.R. 1.98(a)(2), claim 1 stands rejected under 35 U.S.C.§102(e) as being anticipated by Kubler et al. (U.S. Patent No. 5,726,984); and that claims 2-12 stand rejected under 35 U.S.C.§103(a) as being unpatentable over the '984 reference.

Claim 12 has been amended and new claims 16 - 23 have been added. Support for the amendment and the new claims may be found, for example, on page 11, lines 4 - 21 of the specification.